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ATTORNEY'S DOCKET NO.
066303.0112

PATENT APPLICATION

1

APPLICATION FOR U.S. PATENT UNDER 37 C.F.R. § 1.53(b)
TRANSMITTAL FORM

Box Patent Application
ASSISTANT COMMISSIONER FOR PATENTS
Washington, D.C. 20231

Sir:

Transmitted herewith for filing is the patent application
of:

Inventor or Application Identifier:
A. J. Paul Carew, et al.

Entitled: METHOD AND APPARATUS FOR PROVIDING VOICE SIGNALS
TO AND FROM A TELECOMMUNICATIONS SWITCH

Enclosed are:

☒ Specification (24 Total Pages)
☒ Drawing(s) (2 Total Sheet(s) of ☒ Formal ☐ Informal)
☒ Combined Declaration and Power of Attorney (☐ Total Pages)
 Newly executed (original or copy)
☒ Copy from a prior application
 (for continuation/divisional only)

☒ The prior application was assigned of record to General
Bandwidth Inc., a Delaware corporation.

☒ Information Disclosure Statement (IDS) PTO-1449
 Copies of IDS Citations.

☒ Preliminary Amendment

☒ Certificate of Mailing

☒ Return Receipt Postcard

☐ Other _____

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PATENT APPLICATION

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Applicant is:

☐ Large Entity

☒ Small Entity

☐ Small Entity Statement enclosed

☒ Small Entity Statement filed in prior application.

Status still proper and desired.

The accompanying application is:

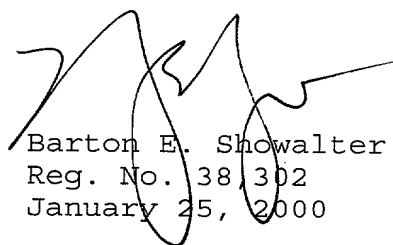
☐ Original

☒ Continuation ☐ Divisional ☐ Continuation-In-Part (CIP)
of prior application No. 09/356,250 which is hereby
incorporated by reference therein.

FEE CALCULATION					FEE
			Number Extra	Rate	Basic Fee
	Number				\$ 345.00
Total Claims:	49	- 20 =	29	X \$9 =	\$ 261.00
Independent Claims	5	- 3 =	2	X \$39 =	\$ 78.00
TOTAL FILING FEE =					\$ 684.00

X Enclosed is a check in the amount of \$684.00 filing fee requirements under 37 C.F.R. § 1.16. Please charge any additional fees or credit any overpayment to Deposit Account No. 02-0384 of BAKER BOTTS L.L.P. A duplicate copy of this sheet is enclosed.

Respectfully submitted,
BAKER & BOTTS, L.L.P.
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Barton E. Showalter
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January 25, 2000

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Attorney's Docket:
066303.0112

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Application of: A. J. Paul Carew, et al.
Date Filed: January 25, 2000
Title: METHOD AND APPARATUS FOR
PROVIDING VOICE SIGNALS TO AND FROM
A TELECOMMUNICATIONS SWITCH

Box Patent Application
Assistant Commissioner for Patents
Washington, D.C. 20231
Dear Sir:

CERTIFICATE OF MAILING BY EXPRESS MAIL

I hereby certify that the attached Application Transmittal Form, Preliminary Amendment, Drawings, Declaration and Power of Attorney (copy from a prior application), Small Entity Statement (copy from a prior application), Information Disclosure Statement, PTO 1449, and firm check in the amount of \$684.00 are being deposited with the United States Postal Service "Express Mail Post Office to Addressee" service under 37 C.F.R. 1.10 on this 25th day of January, 2000, addressed to the Assistant Commissioner for Patents, Washington, D.C. 20231.



Willie Jiles

Express Mail Receipt
No. EM082813984US

Applicant or Patentee: A. J. I. Carew, et al.
Serial or Patent No. _____

Attorney . Docket No. 066303.0105

Filed or Issued: July 16, 1999

Title: METHOD AND APPARATUS FOR PROVIDING VOICE SIGNALS
TO AND FROM A TELECOMMUNICATIONS SWITCH

**STATEMENT CLAIMING SMALL ENTITY STATUS
(37 CFR 1.9(f) & 1.27(c)) - SMALL BUSINESS CONCERN**

I hereby state that I am an official of the small business concern empowered to act on behalf of the concern identified below:

Name of Small Business Concern: General Bandwidth Inc.
Address of Small Business Concern: 8303 N. Mopac, Suite A-101
Austin, Texas 78759

I hereby state that the above-identified small business concern qualifies as a small business concern as defined in 13 CFR 121.12, and reproduced in 37 CFR 1.9(d), for purposes of paying reduced fees to the United States Patent and Trademark Office, in that the number of employees of the concern, including those of its affiliates, does not exceed 500 persons. For purposes of this statement, (1) the number of employees of the business concern is the average over the previous fiscal year of the concern of the persons employed on a full-time, part-time or temporary basis during each of the pay periods of the fiscal year, and (2) concerns are affiliates of each other when either, directly or indirectly, one concern controls or has the power to control the other, or a third party or parties controls or has the power to control both.

I hereby state that rights under contract or law have been conveyed to and remain with the small business concern identified above with regard to the invention, entitled METHOD AND APPARATUS FOR PROVIDING VOICE SIGNALS TO AND FROM A TELECOMMUNICATIONS SWITCH by inventor(s) A. J. Paul Carew, Ronald D. Lutz, Jr., and Brendon W. Mills, described in the specification filed herewith.

If the rights held by the above-identified small business concern are not exclusive, each individual, concern or organization having rights in the invention is listed below, and no rights to the invention are held by any person, other than the inventor, who would not qualify as an independent inventor under 37 CFR 1.9(c) if that person made the invention, or by any concern which would not qualify as a small business concern under 37 CFR 1.9(d), or a nonprofit organization under 37 CFR 1.9(e):

NONE

I acknowledge the duty to file, in this application or patent, notification of any change in status resulting in loss of entitlement to small entity status prior to paying, or at the time of paying, the earliest of the issue fee or my maintenance fee due after the date on which status as a small entity is no longer appropriate. (37 CFR 1.28(b)).

Name of Person Signing: Brendon W. Mills
Title of Person if other than owner: President and CEO
Address of Person Signing: 8303 N. Mopac, Suite A-101
Austin, Texas 78759

Signature: 

Date: 7-15-99

ATTORNEY'S DOCKET NO.
066303.0112

PATENT APPLICATION

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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re application of: A. J. PAUL CAREW, ET AL.

For: METHOD AND APPARATUS FOR PROVIDING VOICE
SIGNALS TO AND FROM A TELECOMMUNICATIONS
SWITCH

I hereby certify that this communication is being deposited with the United States Postal Service "Express Mail Post Office to Addressee" under 37 C.F.R. 1.10 and is addressed to: Assistant Commissioner for Patents, Washington, D.C. 20231, on January 25, 2000.

Willie Jiles

Willie Jiles

Assistant Commissioner

for Patents

Washington, D.C. 20231

Express Mail Cert. No.
EM082813984US

PRELIMINARY AMENDMENT

Prior to the initial review of this continuation application of Application Serial No. 09/356,250 filed July 16, 1999, by A. J. Paul Carew, Ronald D. Lutz, Jr., and Brendon W. Mills, entitled "Method and Apparatus for Providing Voice Signals to and from a Telecommunications Switch," please amend the application as follows:

IN THE SPECIFICATION

Page 1, line 1, please insert the following:

-- CROSS-REFERENCE TO RELATED APPLICATION:

This application is a continuation of U.S. Application Serial No. 09/356,250, filed July 16, 1999, by A. J. Paul Carew, Ronald D. Lutz, Jr., and Brendon W. Mills and entitled "Method and Apparatus for Providing Voice Signals to and from a Telecommunications Switch," pending.--

IN THE CLAIMS

For the convenience of the Examiner, all pending claims are shown below whether or not an amendment has been made. Please amend the claims as follows:

Please cancel Claims 1-35 and add the following new claims therefor:

36. (New) A system for supporting oversubscription, comprising:
a telecommunications switch operable to assign a plurality of telephone numbers to a line, to receive an incoming call for one of the telephone numbers, and to communicate the incoming call associated with the telephone number using the line; and
a voice gateway coupled to the telecommunications switch using the line, the voice gateway operable to receive the incoming call, to detect a distinctive ring, and to communicate the incoming call according to the distinctive ring.
37. (New) The system of Claim 36, wherein the voice gateway communicates the incoming call by processing the incoming call into digital packets according to the distinctive ring and communicating the digital packets to a customer premises interface for further communication to a customer premises.
38. (New) The system of Claim 37, wherein the voice gateway processes the incoming call into the digital packets according to the distinctive ring by identifying an address associated with the distinctive ring and assigning the address to the digital packets.

39. (New) The system of Claim 38, wherein the address is an Internet Protocol (IP), Asynchronous Transfer Mode (ATM), or Frame Relay address.

40. (New) The system of Claim 37, wherein the customer premises interface is a Digital Subscriber Line Access Multiplexer (DSLAM) operable to communicate the digital packets over a twisted pair in a local loop using a digital subscriber line.

41. (New) The system of Claim 36, wherein the voice gateway is further operable to communicate the incoming call to a selected one of a plurality of output lines according to the distinctive ring.

42. (New) The system of Claim 36, wherein the telecommunications switch is further operable to assign at least four telephone numbers to the line.

43. (New) The system of Claim 36, wherein the voice gateway is further operable to receive an outgoing call originated at a customer premises, to identify an available line from a plurality of lines coupled between the telecommunications switch and the voice gateway, and to communicate the outgoing call to the telecommunications switch using the available line.

44. (New) The system of Claim 43, wherein the voice gateway communicates the outgoing call by receiving digital packets, processing the digital packets into a voice signal, and communicating the voice signal to the telecommunications switch using the available line.

45. (New) The system of Claim 43, wherein the plurality of lines is a hunt group.

46. (New) The system of Claim 36, wherein:
the telecommunications switch is a Class 5 switch; and
the voice gateway is further operable to couple to the Class 5 switch without using an
overlay Class 5 switch or digital loop carrier architecture.

47. (New) The system of Claim 36, wherein the line is an unbundled analog line.

48. (New) A voice gateway for supporting oversubscription of a line coupled to a telecommunications switch, the voice gateway operable to receive a first incoming call with a first distinctive ring from the line and to communicate the first incoming call to a first destination according to the first distinctive ring, the voice gateway further operable to receive a second incoming call with a second distinctive ring from the line and to communicate the second incoming call to a second destination according to the second distinctive ring.

49. (New) The voice gateway of Claim 48, wherein the voice gateway communicates the first incoming call by processing the first incoming call into digital packets according to the first distinctive ring and communicating the digital packets to a customer premises.

50. (New) The voice gateway of Claim 49, wherein the voice gateway processes the first incoming call into the digital packets according to the first distinctive ring by identifying an address associated with the first distinctive ring and assigning the address to the digital packets.

51. (New) The voice gateway of Claim 50, wherein the address is an Internet Protocol (IP), Asynchronous Transfer Mode (ATM), or Frame Relay address.

52. (New) The voice gateway of Claim 49, wherein the voice gateway communicates the digital packets to the customer premises using a Digital Subscriber Line Access Multiplexer (DSLAM) operable to communicate the digital packets over a twisted pair in a local loop using a digital subscriber line.

53. (New) The voice gateway of Claim 48, wherein the voice gateway is further operable to communicate the first incoming call to a selected one of a plurality of output lines according to the first distinctive ring.

54. (New) The voice gateway of Claim 48, wherein the voice gateway receives the second incoming call after terminating the first incoming call.

55. (New) The voice gateway of Claim 48, wherein the voice gateway is further operable to support oversubscription of at least 4:1.

56. (New) The voice gateway of Claim 48, wherein the voice gateway is further operable to receive an outgoing call originated at a customer premises, to identify an available line from a plurality of lines coupled to the telecommunications switch, and to communicate the outgoing call to the telecommunications switch using the available line.

57. (New) The voice gateway of Claim 56, wherein the voice gateway communicates the outgoing call by receiving digital packets, processing the digital packets into a voice signal, and communicating the voice signal to the telecommunications switch using the available line.

58. (New) The voice gateway of Claim 56, wherein the plurality of lines is a hunt group.

59. (New) The voice gateway of Claim 48, wherein:
the telecommunications switch is a Class 5 switch; and
the voice gateway is further operable to couple to the Class 5 switch without using an overlay Class 5 switch or digital loop carrier architecture.

60. (New) The voice gateway of Claim 48, wherein the lines are unbundled analog lines.

61. (New) A method for supporting oversubscription of a line coupled to a telecommunications switch, comprising:

receiving a first incoming call with a first distinctive ring from the line coupled to the telecommunication switch;

communicating the first incoming call to a first destination according to the first distinctive ring;

receiving a second incoming call with a second distinctive ring from the line; and

communicating the second incoming call to a second destination according to the second distinctive ring.

62. (New) The method of Claim 61, wherein communicating the first incoming call to the first destination according to the first distinctive ring further comprises:

processing the first incoming call into digital packets according to the first distinctive ring; and

communicating the digital packets to a customer premises.

63. (New) The method of Claim 62, wherein processing the first incoming call into the digital packets according to the first distinctive ring further comprises:

identifying an address associated with the first distinctive ring; and

assigning the address to the digital packets.

64. (New) The method of Claim 63, wherein the address is an Internet Protocol (IP), Asynchronous Transfer Mode (ATM), or Frame Relay address.

65. (New) The method of Claim 62, wherein communicating the digital packets to the customer premises further comprises communicating the digital packets to a customer premises interface for further communications to the customer premises.

66. (New) The method of Claim 65, wherein the customer premises interface is a Digital Subscriber Line Access Multiplexer (DSLAM) operable to communicate the digital packets over a twisted pair in a local loop using a digital subscriber line.

67. (New) The method of Claim 61, wherein communicating the first incoming call to the first destination according to the first distinctive ring further comprises:
selecting one of a plurality of output lines according to the first distinctive ring; and
communicating the first incoming call using the selected output line.

68. (New) The method of Claim 61, further comprising terminating the first incoming call before receiving the second incoming call.

69. (New) The method of Claim 61, further comprising providing at least 4:1 oversubscription of the line.

70. (New) The method of Claim 61, further comprising:
receiving an outgoing call from a customer premises;
identifying an available line from a plurality of lines coupled to the telecommunications switch; and
communicating the outgoing call to the telecommunications switch using the available line.

71. (New) The method of Claim 70, wherein communicating the outgoing call to the telecommunications switch further comprises:

receiving digital packets from a customer premises interface;

processing the digital packets into a voice signal; and

communicating the voice signal to the telecommunications switch using the available line.

72. (New) The method of Claim 70, wherein the plurality of lines is a hunt group.

73. (New) The method of Claim 61, wherein:

the telecommunications switch is a Class 5 switch; and

the lines couple to the Class 5 switch without using an overlay Class 5 switch or digital loop carrier architecture.

74. (New) The method of Claim 61, wherein the lines are unbundled analog lines.

75. (New) A voice gateway for supporting oversubscription of a plurality of unbundled lines coupled to a telecommunications switch, the voice gateway operable to receive digital packets from a customer premises, to process the digital packets into a voice signal, to identify an available unbundled line from the plurality of unbundled lines, and to communicate the voice signal to the telecommunications switch using the available unbundled line.

76. (New) The voice gateway of Claim 75, wherein the voice gateway receives the digital packets from the customer premises using a Digital Subscriber Line Access Multiplexer (DSLAM) operable to receive digital packets over a twisted pair in a local loop using a digital subscriber line.

77. (New) The voice gateway of Claim 75, wherein the plurality of unbundled lines is a hunt group.

78. (New) The voice gateway of Claim 75, wherein the unbundled lines are Integrated Services Digital Network Basic Rate Interface (ISDN BRI) lines, each ISDN BRI line operable to simultaneously communicate two voice signals between the telecommunications switch and the voice gateway.

79. (New) The voice gateway of Claim 75, wherein:
the telecommunications switch is a Class 5 switch; and
the unbundled lines couple to the Class 5 switch without using an overlay Class 5 switch or digital loop carrier architecture.

80. (New) A method for supporting oversubscription of a plurality of unbundled lines coupled to a telecommunications switch, further comprising:
receiving digital packets from a customer premises;
processing the digital packets into a voice signal
identifying an available unbundled line from the plurality of unbundled lines; and
communicating the voice signal to the telecommunications switch using the available unbundled line.

81. (New) The method of Claim 80, wherein receiving the digital packets from the customer premises further comprises using a Digital Subscriber Line Access Multiplexer (DSLAM) to receive digital packets over a twisted pair in a local loop using a digital subscriber line.

82. (New) The method of Claim 80, wherein the plurality of unbundled lines is a hunt group.

83. (New) The method of Claim 80, wherein the unbundled lines are Integrated Services Digital Network Basic Rate Interface (ISDN BRI) lines, each ISDN BRI line operable to simultaneously communicate two voice signals between the telecommunications switch and the voice gateway.

84. (New) The method of Claim 80, wherein:
the telecommunications switch is a Class 5 switch; and
the unbundled lines couple to the Class 5 switch without using an overlay Class 5 switch or digital loop carrier architecture.

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PATENT APPLICATION

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REMARKS

Early and favorable acceptance of this continuation application is respectfully requested.

Respectfully submitted,
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METHOD AND APPARATUS FOR PROVIDING VOICE SIGNALS
TO AND FROM A TELECOMMUNICATIONS SWITCH

TECHNICAL FIELD OF THE INVENTION

The present invention relates in general to telecommunications signal processing and more particularly to a method and apparatus for providing voice signals to and from a telecommunications switch.

5

BACKGROUND OF THE INVENTION

The traditional circuit switched telecommunications network has been implemented to dedicate one voice line to one loop or copper pair. This has worked well for over a hundred years but does not efficiently utilize the bandwidth of the copper pair. In addition, there has been a surge in demand for second, and even third, residential phone lines. This demand is exhausting the supply of available copper circuits. Business customers also have a high demand for phone lines. To meet this demand, Regional Bell Operating Companies, Independent Local Exchange Carriers, and Competitive Local Exchange Carriers would have to build additional copper or fiber infrastructure.

New technology, such as Digital Subscriber Line, voice-over-IP, and asynchronous transfer mode techniques have created an environment where the copper pair's available bandwidth can be more fully utilized to carry voice and data. However, traditional voice traffic is time division multiplexed, a transport architecture that segments the network bandwidth into fixed time sequenced channels. The smallest channel is equivalent to a voice line. Time division multiplexed networks work well for uncompressed analog voice but not for bursty data. If a data network needs more than 64 kilobits per second of bandwidth, the amount of one channel, two channels would be needed to carry 65 kilobits per second, resulting in bandwidth inefficiencies.

With the explosion of the Internet, worldwide deployment of Digital Subscriber Lines will rapidly accelerate over the next few years. Today, however, the penetration rate for voice over DSL is at zero percent. With the increase in their deployment, DSL is a prime candidate for implementing a multiple voice line capability for telecommunications customers. There have been recent efforts to provide voice over DSL. However, these efforts

have required a GR-303 connection with a Class 5 switch for the gateway device. This GR-303 connection is available at the regional bell operating company or independent local exchange carrier level but competitive local exchange carriers would need to provide there own Class 5 switch or digital loop carrier functionality to interface with the GR-303 connection. In order to implement this functionality, competitive local exchange carriers would have to incur costly expense in providing this infrastructure. Therefore, it is desirable to migrate voice services into the data transport network in order to efficiently use the bandwidth of the copper pair and avoid expensive infrastructure changes in allowing a competitive local exchange carrier to implement an increased and efficient voice transport capability.

SUMMARY OF THE INVENTION

From the foregoing, it may be appreciated that a need has arisen to efficiently provide voice signal transport without bandwidth inefficiency. In accordance with the present invention, a method and apparatus for providing voice signals to and from a telecommunications switch are provided which substantially eliminate or reduce disadvantages and problems associated with conventional voice transport techniques.

According to an embodiment of the present invention, there is provided an apparatus for providing voice signals from a telecommunications switch that includes an input port operable to receive an unbundled analog line from the telecommunications switch, wherein a voice signal is carried over the analog line. An analog-to-digital converter unit converts the voice signal carried on the analog line into a digital format. A compressing unit places the voice signal into a compressed format. A packetizing unit places the voice signal into a packet format for transport over a data network.

The present invention provides various technical advantages over conventional voice transport techniques. For example, one technical advantage is to provide unbundled analog line ports to a competitive local exchange carrier without the need for an overlay Class 5 switch or digital loop carrier architecture. Another technical advantage is to mimic the dynamic allocation of timeslots of a standard GR-303 interface without utilizing that interface in order to provide an oversubscription capability. Yet another technical advantage is the ability to support a multitude of voice transport framing philosophies. Still another technical advantage is to provide selective compression and packetizing capabilities for versatile voice transport operation. Other technical

advantages may be readily apparent to those skilled in the art from the following figures, description, and claims.

BRIEF DESCRIPTION OF THE DRAWINGS

For a more complete understanding of the present invention and the advantages thereof, reference is now made to the following description taken in conjunction with the accompanying drawings, wherein like reference numerals represent like parts, in which:

FIGURE 1 illustrates a block diagram of a telecommunications network;

FIGURE 2 illustrates a block diagram of a voice gateway within the telecommunications network.

DETAILED DESCRIPTION OF THE INVENTION

FIGURE 1 is a block diagram of a portion of a telecommunications network 10. Telecommunications network 10 includes one or more office customer premises 12, one or more residence customer premises 14, one or more information service providers 16, one or more independent local exchange carrier central offices 18, and one or more competitive local exchange carriers 20.

Office customer premises 12 may receive data and voice at an Integrated Access Device (IAD) 22. IAD 22 may provide data and voice to a private branch exchange 24 in order to support telephony operations at telephony devices 25 within office customer premises 12. IAD 22 may also provide data and voice to a local area network 26 through a router 27. Local area network 26 may have computers or other devices 28 connected thereto for processing the data and voice received from IAD 22 in order to support computer processing and telephony capability over local area network 26. Data and voice generated by devices 25 and 28 connected to local area network and private branch exchange 24 may also be transferred out of office customer premises 12 by IAD 22 .

Residence customer premises 14 may receive data and voice at an IAD 30. IAD 30 may provide data and voice to telephony devices 32 and also to computing devices 34 connected thereto. Data and voice generated by either telephony devices 32 or computing devices 34 or both may be transferred out of residence customer premises 14 through IAD 30.

Information service provider 16 may receive data at an Internet gateway 40 from competitive local exchange carrier 20. Internet gateway 40 provides the interface to Internet 42. Information service provider 16 supports connections to Internet 42 for the passage of data thereto and

therefrom through Internet gateway 40 as received from or provided to competitive local exchange carrier 20.

Independent local exchange carrier central office 18 may receive data and voice carried by a public switched telephone network 50. A Class 5 switch 52 is the interface to and from public switched telephone network 50. Class 5 switch 52 passes voice and data received from public switched telephone network 50 to competitive local exchange carrier 20. Competitive local exchange carrier 20 provides voice and data to office customer premises 12 and residence customer premises 14 from Class 5 switch 52. Voice and data may be received from office customer premises 12 and residence customer premises 14 by competitive local exchange carrier 20 for transfer to Class 5 switch 52.

Competitive local exchange carrier 20 includes a voice gateway 64 receives voice and data from and provides voice and data Class 5 switch 52. Unbundled analog line connections 62 are provided from Class 5 switch 52 to voice gateway 64. By providing a capability to interconnect to Class 5 switch 52 using standard unbundled analog lines 62, competitive local exchange carrier 20 is able to provide voice functionality over its broadband network without needing its own overlay Class 5 switch or digital loop carrier architecture. A Digital Subscriber Line Access Multiplexer (DSLAM) device 66 provides an interface for voice and data with office customer premises 12 and residence customer premises 14. DSLAM device 66 and voice gateway 64 pass voice and data to and from each other or to and from information service provider 16 through a packet switch 68. Packet switch 68 may operate using any of a variety of packet techniques to include asynchronous transfer mode and frame relay. Voice and data may be transferred throughout telecommunications network 10 in any of a variety of packet formats to include asynchronous transfer mode cells, frame relay packets, and Internet

protocol. Competitive local exchange carrier 20 may also implement multiple packet switches 68, each using a different packet technique. Though shown, competitive local exchange carrier 20 need not be colocated with independent local exchange carrier central office 18.

For voice operation from public switched telephone network 50, a voice signal is transferred over public switched telephone network 50 to Class 5 switch 52. Class 5 switch 52 routes the voice signal to voice gateway 64 over an unbundled analog line 62. Voice gateway processes the voice signal for transfer to DSLAM device 66 through packet switch 68. The processing that may be performed by voice gateway 64 may include multiplexing, analog-to-digital conversion, compression, and packetizing. DSLAM device 66 provides the voice signal to its intended destination, such as office customer premises 12 or residence customer premises 14.

For voice operation to public switched telephone network 50, a voice signal is generated at, for example, office customer premises 12 and transferred to DSLAM device 66. DSLAM device 66 receives the voice signal from office customer premises 12 and prepares the voice signal for transport over packet switch 68 to voice gateway 64. Upon receipt of the voice signal, voice gateway 64 converts the voice signal into its appropriate analog format for transfer over an unbundled analog line 62. The unbundled analog line 62 carries the voice signal to Class 5 switch 52. Class 5 switch 52 transfers the voice signal to its appropriate destination on public switched telephone network 50.

Voice gateway 64 provides a capability to packetize and compress circuit switched voice circuits from public switched telephone network 50 and deliver them over broadband networks to business and residential customers. With this capability, telecommunications service providers

may offer, according to the preferred embodiment, twenty-four independent voice lines over one Digital Subscriber Line circuit. Voice gateway 64 supports a variety of types of network framing, such as voice over asynchronous transfer mode and voice over Internet protocol. Additionally, voice gateway 64 supports the latest in voice compression technologies so that calls placed through voice gateway 64 sound similar to calls placed over public switched telephone network 50. Multiple phone lines can be imbedded into the broadband data stream and additional lines can be added or subtracted on demand over a single copper circuit.

FIGURE 2 is a block diagram of voice gateway 64. Voice gateway 64 includes an input port 70 to receive and provide voice signals from and to unbundled analog lines 62 of Class 5 switch 52. An analog-to-digital and digital-to-analog converter unit 76 includes A/D and D/A converters 78 that convert voice signals received from unbundled analog lines 62 into a digital format and convert voice signals transferred to unbundled analog lines 62 into an analog format. A compressing/decompressing unit 80 includes compressors/decompressors 82 that compress voice signals received from A/D and D/A converters 78 into a compressed format and decompress voice signals prior to conversion into analog format. A packetizing unit 84 includes packetizers/depacketizers 86 that packetize voice signals into a packet format and depacket voice signals from the packet format prior to decompression. An output port 88 provides voice signals to and receives voice signals from DSLAM device 66 through packet switch 68. Output port 88 is capable of multiplexing multiple voice signals together through interleaving of packets of different voice signals onto an output line to packet switch 68. Output port 88 may also selectively place any voice signal onto any of its output lines according to the destination characteristics

of each voice signal. For example, output port may multiplex five voice signals onto a first output line and multiplex three other voice signals onto a second output line.

5 Compressors/decompressors 82 may implement different compression ratios. For example, compressor/decompressor R1 may perform voice compression using the standard G.711 compression technique of 64 kilobits per second pulse code modulation. Compressor/decompressor R2 may perform voice
10 compression using the standard G.722 compression technique of 32 kilobits per second adaptive differential pulse code modulation. Compressor/decompressor R3 may perform voice compression using the standard G.726 compression technique of 16 kilobits per second compression. Whichever
15 compression technique is selected for a voice signal, a customer experience of placing a call through voice gateway 64 will be indistinguishable from a call placed only over public switched telephone network 50. Selection of which compression technique to perform on a particular voice
20 signal is determined by the configuration of a first switching matrix 90. First switching matrix 90 is capable of dynamically routing any voice signal received to any one of compressors/decompressors 82 in order to support selective compression of voice signals. Appropriate
25 decompression is also performed followed by selective routing through first switching matrix 90 to an appropriate A/D and D/A converter 78.

 Packetizers/depacketizers 86 may implement different transport framing philosophies. For example,
30 packetizer/depacketizer P1 may packetize the voice signal into asynchronous transfer mode cells. Packetizer/depacketizer P2 may packetize the voice signal into frame relay packets. Packetizer/depacketizer P3 may packetize the voice signal into an Internet protocol
35 format. The Internet protocol format may then be carried

in the asynchronous transfer mode or frame relay format. Selection of which packetizing technique to perform on a particular voice signal is determined by the configuration of a second switching matrix 92. Second switching matrix 92 is capable of dynamically routing any voice signal received to any one of packetizers/depacketizers 86 in order to support selective packetizing of voice signals. Appropriate depacketizing is also performed followed by selective routing through second switching matrix 92 to an appropriate compressor/decompressor 82.

For voice operation from Class 5 switch 52, a voice signal carried over an associated unbundled analog line 62 is received at input port 70. Input port 70 performs electrical analog termination of the incoming unbundled analog line 62 and insures that the lines are properly terminated. Input port 70 provides the voice signal to a corresponding A/D and D/A converter 76 in analog-to-digital and digital-to-analog converter unit 74. Analog-to-digital and digital-to-analog converter unit 74 may also perform coding and decoding functions of a conventional CODEC unit to include a ring and digit detection unit 75. A control processor 77 may be part of analog-to-digital and digital-to-analog converter unit 74 to supervise and control CODEC functionality. A distinctive ring detection may also be employed to provide an oversubscription capability discussed in more detail later. Analog-to-digital and digital-to-analog converter unit 74 detects a ring condition on unbundled analog line 62 and the corresponding A/D and D/A converter 78 places the voice signal into a digital format. The digitized voice signal passes through first switching matrix 90 where it is routed to a desired compressor/decompressor 82 in compressing/decompressing unit 80. The voice signal is compressed and passes through second switching matrix 92 where it is routed to a desired packetizer/depaketizer 86 in packetizing/depaketizing

unit 84. The voice signal in its packet format transfers through output port 88, possibly multiplexed with other packetized voice signals, and is passed to DSLAM device 66 over an appropriate output line through packet switch 68 for ultimate delivery to office customer premises 12 or residence customer premises 14 over associated digital subscriber lines.

For voice operation to Class 5 switch 52, the voice signal is originated at office customer premises 12 or residence customer premises 14, passes through DSLAM device 66 and packet switch 68, and is received at output port 88 of voice gateway 64. Output port 88 demultiplexes the voice signal provides the voice signal to an associated packetizer/depacketizer 86 in packetizing/depacketizing unit 84 according to an available unbundled analog line 62. The packetizer/depacketizer 86 removes the voice signal from its transport frame. The voice signal then passes through second switching matrix 92 for routing to an appropriate compressor/decompressor 82 in compressing/decompressing unit 80. Compressor/decompressor 82 decompresses the voice signal into its full digital format. The voice signal then passes through first switching matrix 90 for routing to an appropriate A/D and D/A converter 78 in analog-to-digital and digital-to-analog converting unit 76. A/D and D/A converter 78 places the voice signal into its analog format. The voice signal is then placed onto its corresponding unbundled analog line 62 at input port 70. The voice signal then passes on to Class 5 switch 52 for further routing through public switched telephone network 50.

Voice gateway 64 may also support an oversubscription capability. Each unbundled analog line 62 may be provisioned to carry voice traffic in a 1:1 ratio where there are the same number of unbundled analog lines 62 for each telephone number. Each unbundled analog line 62 may

also be oversubscribed, for example in a 4:1 ratio, where there are four telephone numbers per unbundled analog line 62. Unbundled analog line 62 may also be an Integrated Services Digital Network Basic Rate Interface line with a capability to transfer two simultaneous voice channels. The use of this type of line allows for the immediate doubling of call capacity with or without oversubscription.

For outgoing calls toward Class 5 switch 52, a first telephone device associated with a first one of the four telephone numbers may be in use and thus occupying its associated unbundled analog line 62. A second telephone device associated with a second one of the telephone numbers may be put into use as long as there is a free unbundled analog line 62 connected to voice gateway 64. Output port 88 determines if there is a free unbundled analog line 62 available for connection of the second telephone device, such as through a hunt group search. Output port 88 is capable of connecting any telephone device of office customer premises 12 and residence customer premises 14 to any available unbundled analog line 62 in order to support the oversubscription capability.

For incoming calls from Class 5 switch 52, voice gateway 64 is capable of detecting a distinctive ring given to each telephone number assigned to unbundled analog line 62. In the 4:1 oversubscription example, each of the four telephone numbers has its own unique ring associated therewith. Voice gateway 64 determines which of the customer telephone devices to route the call to according to the detected ring. Though described with reference to a 4:1 ratio, other oversubscription ratios may be equally implemented through this technique.

Thus, it is apparent that there has been provided, in accordance with the present invention, a method and apparatus for providing voice signals to and from a telecommunications switch that satisfies the advantages set

forth above. Although the present invention has been described in detail, it should be understood that various changes, substitutions, and alterations may be readily ascertainable by those skilled in the art and may be made
5 herein without departing from the spirit and scope of the present invention as defined by the following claims.

WHAT IS CLAIMED IS:

1. An apparatus for providing voice signals from a telecommunications switch, comprising:

an input port operable to receive an unbundled analog line from the telecommunications switch, the analog line carrying a voice signal;

an analog-to-digital converter unit operable to convert the voice signal carried on the analog line into a digital format;

a compressing unit operable to place the voice signal into a compressed format;

a packetizing unit operable to place the voice signal into a packet format.

2. The apparatus of Claim 1, further comprising:

a switching matrix operable to route the voice signal from the analog-to-digital converter unit to one of a plurality of compressors in the compressing unit.

3. The apparatus of Claim 2, wherein each of the plurality of compressors in the compressing unit performs a different compression type.

4. The apparatus of Claim 3, wherein one of the different compression types is 64 kilobits per second pulse code modulation.

5. The apparatus of Claim 3, wherein one of the different compression types is 32 kilobits per second adaptive differential pulse code modulation.

6. The apparatus of Claim 3, wherein one of the different compression types is 16 kilobits per second compression.

7. The apparatus of Claim 1, further comprising:

a switching matrix operable to route the voice signal from the compressing unit to one of a plurality of packetizers within the packetizing unit.

5

8. The apparatus of Claim 7, wherein each packetizer in the packetizing unit places the voice signal into a different packet format.

10

9. The apparatus of Claim 8, wherein one of the different packet formats is asynchronous transfer mode.

10. The apparatus of Claim 8, wherein one of the different packet formats is Internet Protocol.

15

11. The apparatus of Claim 8, wherein one of the different packet formats is frame relay.

12. The apparatus of Claim 1, further comprising:

20

an output multiplexer operable to receive a plurality of voice signals from the packetizing unit, the output multiplexer operable to selectively multiplex different voice signals together for output onto one of a plurality of output lines.

25

13. The apparatus of Claim 1, wherein the analog-to-digital converter unit includes a plurality of analog-to-digital converters, each analog-to-digital converter operable to receive one of a plurality of unbundled analog lines received at the input port from the telecommunications switch, each analog-to-digital converter operable to place a voice signal carried over its associated unbundled analog line into a digital format.

30

14. The apparatus of Claim 13, further comprising:

a first switching matrix operable to route voice signals from any of the plurality of analog-to-digital converters to any of a plurality of compressors in the compressing unit, each compressor operable to place voice signals into different compressed formats.

15. The apparatus of Claim 14, further comprising:

a second switching matrix operable to route voice signals from any of the plurality of compressors to any of a plurality of packetizers in the packetizing unit, each packetizer operable to place voice signals into a different packet format.

16. The apparatus of Claim 1, wherein the analog-to-digital converter unit includes a ring detection unit, the ring detection unit operable to determine that a voice signal is present on the unbundled analog line.

17. The apparatus of Claim 15, wherein the voice signal is accompanied by a distinctive ring, the distinctive ring being associated with one of a plurality of telephone numbers assigned to the unbundled analog line, the ring detection unit operable to identify the distinctive ring and determine which one of a plurality of subscriber units the voice signal is to be routed.

18. The apparatus of Claim 17, wherein four telephone numbers are assigned to the unbundled analog line.

19. The apparatus of Claim 17, wherein the unbundled analog line is an integrated services digital network basic rate interface line wherein two voice channels are carried simultaneously.

20. The apparatus of Claim 17, wherein voice signals of a second one of the plurality telephone numbers is carried on an available unbundled analog line when the voice signals of the first one of the plurality of telephone numbers is being carried on the unbundled analog line.

21. A method of providing voice signals from a telecommunications switch, comprising:

receiving an unbundled analog line from the telecommunications switch, the analog line carrying a voice signal;

converting the voice signal carried on the analog line into a digital format;

placing the voice signal into a compressed format;

placing the voice signal into a packet format.

22. The method of Claim 21, further comprising:

selecting one of a plurality of compression techniques in order to place the voice signal into the compressed format.

23. The method of Claim 22, wherein one of the plurality of compression techniques is 64 kilobits per second pulse code modulation.

24. The method of Claim 22, wherein one of the plurality of compression techniques is 32 kilobits per second adaptive differential pulse code modulation.

25. The method of Claim 22, wherein one of the plurality of compression techniques is 16 kilobits per second compression.

26. The method of Claim 21, further comprising:
selecting one of a plurality of packetizing techniques
to place the voice signal into the packet format.

5 27. The method of Claim 26, wherein one of the
different packetizing techniques is asynchronous transfer
mode.

10 28. The method of Claim 26, wherein one of the
different packetizing techniques is Internet Protocol.

29. The method of Claim 26, wherein one of the
different packetizing techniques is frame relay.

15 30. The method of Claim 21, further comprising:
multiplexing the voice signal with other voice
signals.

20 31. The method of Claim 30, wherein the multiplexing
step includes interleaving packets of the voice signal with
packets of other voice signals.

25 32. The method of Claim 30, further comprising:
selecting one of a plurality of output lines to
transfer the voice signal.

30 33. The method of Claim 20, further comprising:
identifying a ring signal accompanying the voice
signal.

34. The method of Claim 33, further comprising:
determining a destination for the voice signal in
response to the ring signal.

35. The method of Claim 34, wherein the ring signal is one of a plurality of distinctive rings, each distinctive ring associated with a telephone number assigned to the unbundled analog line.

METHOD AND APPARATUS FOR PROVIDING VOICE SIGNALS
TO AND FROM A TELECOMMUNICATIONS SWITCH

ABSTRACT OF THE DISCLOSURE

5 A voice gateway (64) includes an input port (70) that receives a voice signal from an unbundled analog line (62) coupled to a Class 5 switch (52). The voice signal is converted to a digital format by an analog-to-digital and digital-to-analog converting unit (76). The voice signal is placed into a compressed format by a compressing/decompressing unit (80) using a selected one of a plurality of compression ratios. The voice signal is placed into a transport frame by a packetizing/depacketizing unit (84) according to a selected packet format. The voice signal is multiplexed with other voice signals by an output port (88). The output port (88) places the voice signal onto a selected one of a plurality of output lines in order to transport the voice signal in its transport frame to one of an office customer premises (12) and a residence customer premises (14).

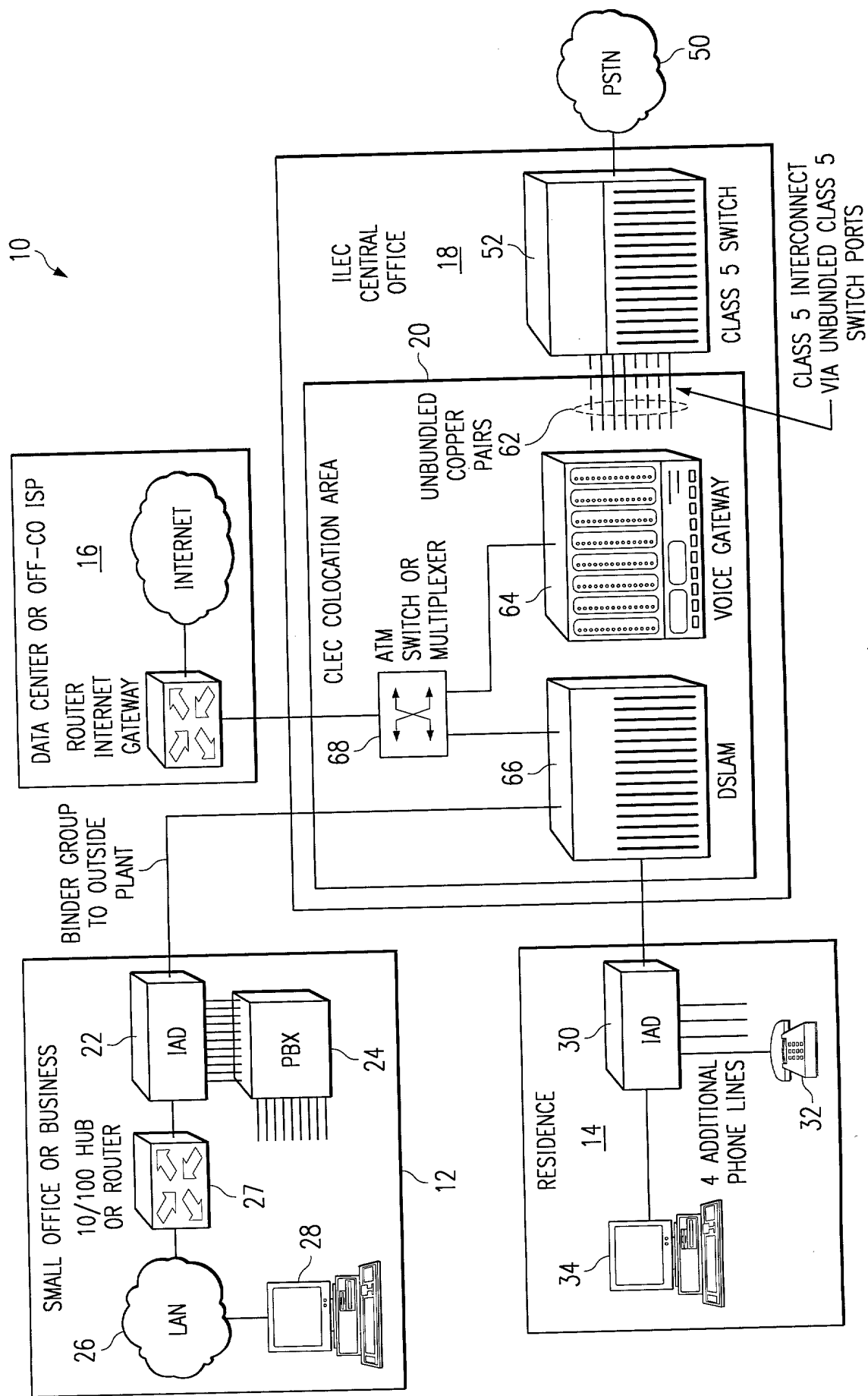


FIG. 1

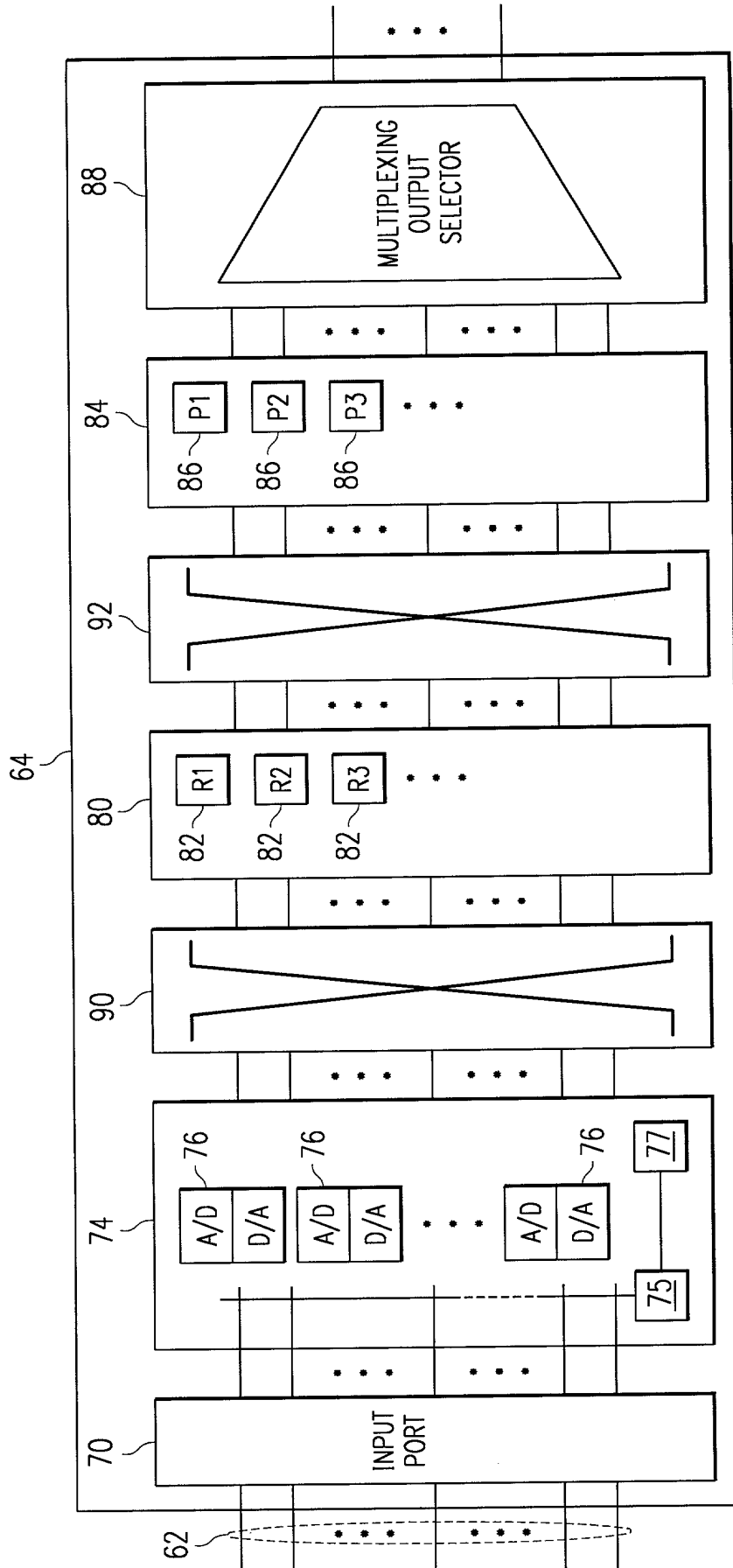
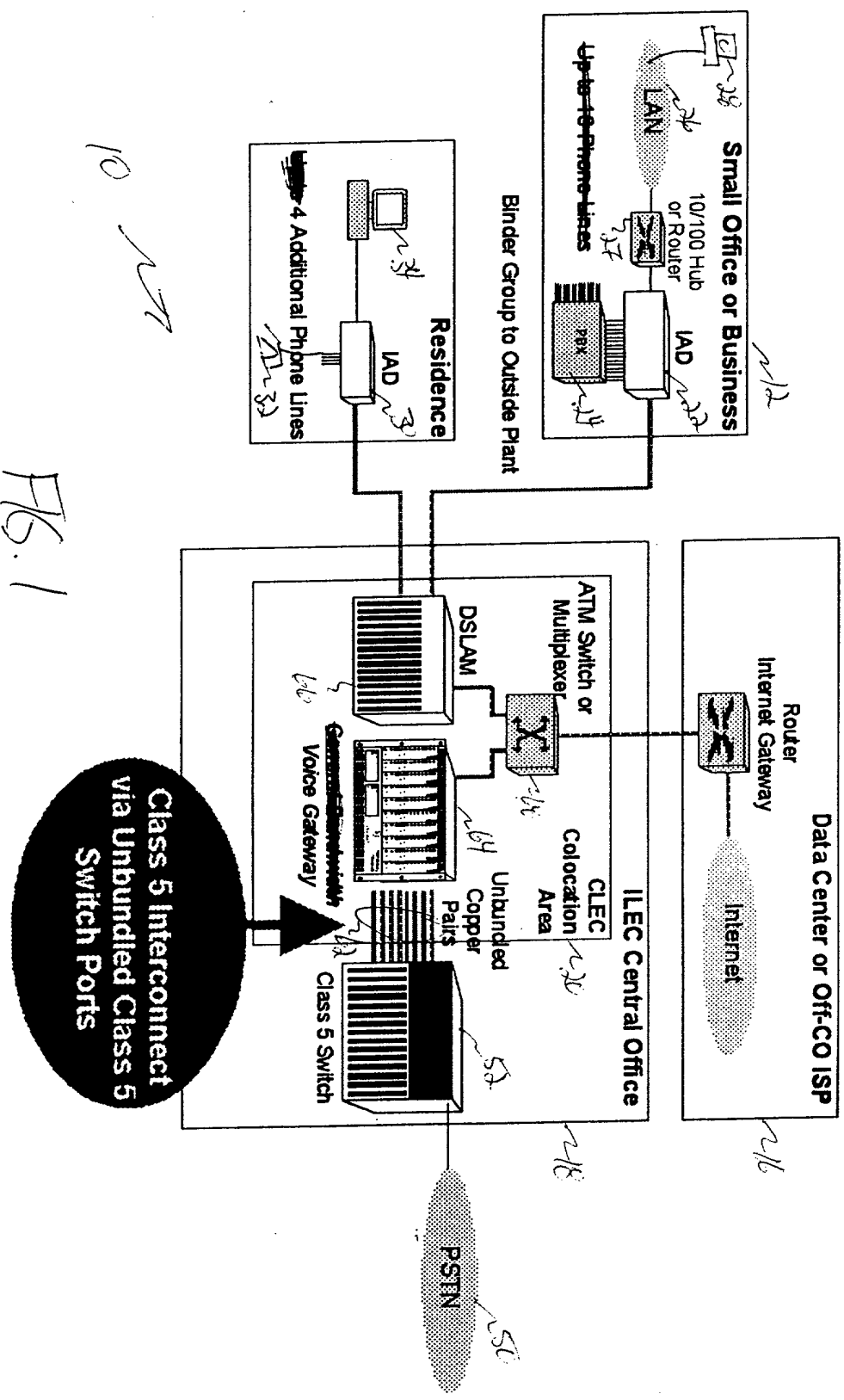
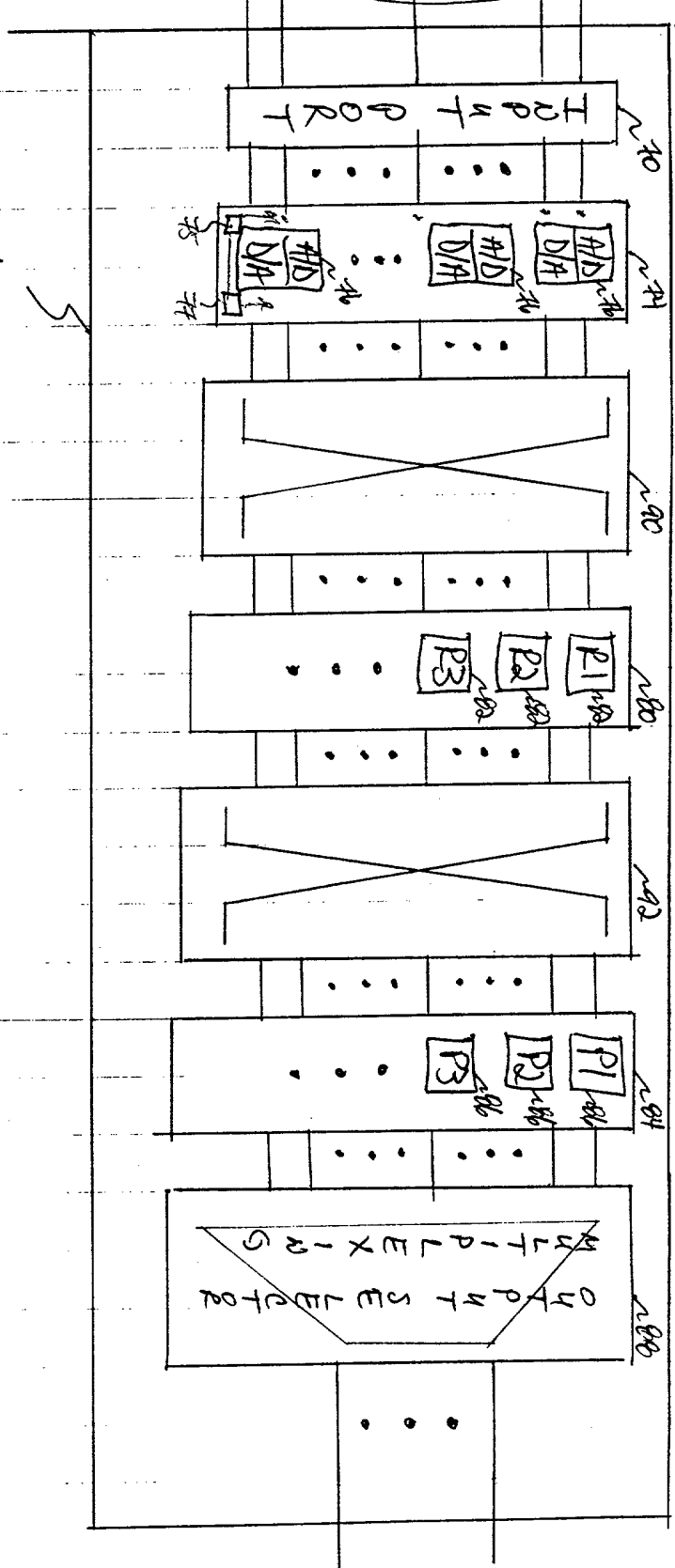
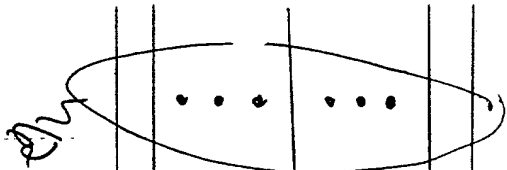


FIG. 2



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64

FS 2

SYSTEM DESIGN

DECLARATION AND POWER OF ATTORNEY

As a below named inventor, I declare:

that my residence, post office address, and citizenship are as stated below next to my name;

that I believe I am the original, first and joint inventor of the subject matter which is claimed and for which a patent is sought on the invention or design entitled METHOD AND APPARATUS FOR PROVIDING VOICE SIGNALS TO AND FROM A TELECOMMUNICATIONS SWITCH, the specification of which (check one):

_____ is attached hereto; or

 X was filed on July 16, 1999 as
Application Serial No. 09/356,250 and was
amended on _____ (if applicable);

that I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above; and that I acknowledge the duty to disclose to the U.S. Patent and Trademark Office all information known to me to be material to patentability as defined in 37 C.F.R. § 1.56.

I further declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the application or any patent issuing thereon.

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